IN THE ABSTRACT:

Please amend the abstract as follows:

--In a mobile wireless communication system automatic speech recognition is performed in a distributed manner using a mobile station based near or front end stage which extracts and vector quantizes recognition feature parameters from frames of an utterance and an infrastructure based back or far end stage which reverses the vector quantization to recover the feature parameters and subjects the feature parameters to a Hidden Markov Model (HMM) evaluation to obtain a recognition decision for the utterance. In order to conserve network capacity, the size (Sz) of the codebook used for the vector quantization, and the corresponding number of bits (B) per codebook index B, are adapted on a dialogue-by dialogue basis in relation to the vocabulary size |V| for the dialogue. The adaptation is performed at the front end, accomplishes a tradeoff between expected recognition rate RR and expected bitrate BR by optimizing a metric which is a function of both. ~~In addition to the frame-wise compression of an utterance into a string of code indices (q-string), further "timewise" compression is obtained by run-length coding the string. The data transmitted from the front end to the back end includes the number of bits (B) per codebook value, which also indicates the codebook size (Sz).~~ --